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**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/043,591 | VICKERS ET AL. | |
| | Examiner | Art Unit | |
| | LUN-SEE LAO | 2614 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on *25 July 2008*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-21* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-21* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

## DETAILED ACTION

*Introduction*

1. This action is in response to the amendments filed on 07-25-2008. Claims 1, 2, 14, 18 and 20 have been amended. Claims 1-21 are pending.

***Claim Rejections - 35 USC § 103***

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1-3 and 14-21 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders (US PAT. 6,351,733) in view of Kitamura (US PAT. 6,704,421).

Regarding Claim 1 Saunders discloses a method of adjusting the dynamics of an audio track, comprising:

deriving, from the transfer function (reads on VRA function), a time-varying gain to modify the statistical distribution of levels of the audio track (see Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67); and

applying the time-varying gain to the audio track to obtain a resulting audio track, wherein the transfer function comprises a multi-line compression transfer function (reads on, PCPV/PCA and/or SCRA) having one or more compression thresholds

(reads on, 8:1), and wherein the parameters include the one or more compression thresholds that are derived on from a fractional measure of the audio track at one or more predetermined levels (see Figs. 1, 5, and 13-14; and see column 13, line 8 to column 14, line 10; column 17, line 16 to column 18, line 34; column 23, lines 48-67); but Saunder does not explicitly teach that the parameters include the one or more compression thresholds that are derived on from a fractional measure of a number of frames of the audio track at one or more predetermined levels.

However, Kitamura teaches that the parameters include the one or more compression thresholds that are derived on from a fractional measure (by compared) of a number of frames (reads on, number of samples) of the audio track at one or more predetermined levels (see figs. 1-4 and col.5 line 41-col. 6 line 67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to combine the teaching of Kitamura into Saunders to provide a preferred equalization requirement and automatically adjust equalization of the audio and to provide a proper multichannel audio output for the sound system.

Regarding Claim 2, Saunders discloses that the step of deriving the transfer function comprises:

specifying a desired statistical dynamics distribution (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67; column 22, lines 53-60); and deriving the parameters (see col. 10 line 2-20) of the transfer function(reads on, wave form in each channel) from the metadata and from the desired statistical dynamics distribution (Figs. 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-

67; column 22, lines 53-60); such that a final statistical dynamics distribution encountered in the resulting audio track after application of the time-varying gain is substantially similar to the desired statistical dynamics distribution (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67).

Regarding Claim 3, Saunders discloses deriving the time varying gain comprises: specifying a desired overall loudness for the audio track (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67; column 22, lines 53-67) deriving an estimate of the loudness of the resulting audio track from the metadata and from an initial estimate of the time-varying gain (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67; column 22, lines 53-67); deriving a correction factor from the desired overall loudness and from the estimate of the loudness of the resulting audio track (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67; column 22, lines 53-67; column 23, line 48 to column 24, line 6); and applying the correction factor to the initial estimate of the time-varying gain to obtain the time-varying gain (Figs. 1 and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 57-67; column 22, lines 53-67; column 23, line 48 to column 24, line 6; column 26, line 40 to column 27, line 4).

Regarding Claim 14, Saunders discloses a method of adjusting the loudness of an audio track including a plurality of audio frames, the method comprising:

obtaining loudness values for each of the plurality of audio frames (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34);

applying a weighting factor (see fig.13 (11340) and see col.22 line 53-col. 23 line 67) to each of the loudness values to obtain a plurality of weighted loudness values (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34);

aggregating the weighted loudness values (see fig.13 (multiplier)) to obtained an overall loudness value for the audio track (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67);

comparing the overall loudness value (see col. 17 line 46-67) to a desired loudness value (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67; column 22, lines 53-60; column 26, line 40 to column 4); and

applying a gain (see fig.13, (1340)) to the audio track based on the comparison between the overall loudness value and the desired loudness value (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67; column 22, lines 53-60; column 26, line 40 to column 4); wherein applying the gain comprises compressing with a multi-line compression transfer function derived from statistical frequency data, the multi-line compression transfer function(reads on, PCPV/PCA and/or SCRA) including one or more compression thresholds (reads on, 8:1) that are derived from a fractional measure of the audio track at one or more predetermined levels (see Figs. 1, 5, and 13-14; and see column 13, line 8 to column 14, line 10; column 17, line 16 to column 18, line 34; column 23, lines 48-67); but Saunders does not explicitly teach the multi-line compression transfer function including one or more compression thresholds that are derived from a fractional measure of a number of frames of the audio track at one or more predetermined levels.

However Kitamura teaches that applying the gain comprises compressing with a multi-line compression transfer function (reads on, multichannel) derived from statistical frequency data, the multi-line compression transfer function including one or more compression thresholds that are derived from a fractional measure(by compared) of a number of frames (reads on number of samples) of the audio track at one or more predetermined levels (see figs. 1-4 and col.5 line 41-col. 6 line 67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to combine the teaching of Kitamura into Saunders to provide a preferred equalization requirement and automatically adjust equalization of the audio and to provide a proper multichannel audio output for the sound system.

Regarding Claim 15, Saunders discloses the weighting factor to be applied to a particular loudness value is derived from the particular loudness value itself (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67; column 22, lines 53-60; column 26, line 40 to column 4).

Regarding Claim 16, Saunders discloses the weighting factor for a particular loudness value comprises an emphasis parameter raised to a power of the particular loudness value (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34; column 23, lines 48-67; column 22, lines 53-60; column 26, line 40 to column 4).

Regarding Claim 18, Saunders discloses a method of altering a dynamic range of an audio track comprising a plurality of audio frames each having a loudness value, the method comprising: obtaining original statistical frequency data for the audio track (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34); applying a test compression

scheme to the original statistical frequency data to obtain test statistical frequency data (Figs. 1,2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34); deriving from the original statistical frequency data and the test statistical frequency data an actual compression scheme (Figs. 1, 2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34); and compressing the audio track using the actual compression scheme (Figs. 1, 2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34); wherein compressing using the actual compression scheme comprises compressing with a multi-line compression transfer function(reads on PCPV/PCA and/or SCRA) derived from the statistical frequency data, the multi- line compression transfer function including one or more compression thresholds (reads 8:1) that are derived from a fractional measure of the audio track at one or more predetermined levels (see Figs. 1, 5, and 13-14; and see column 13, line 8 to column 14, line 10; column 17, line 16 to column 18, line 34; column 23, lines 48-67); but Saunders does not explicitly teach the multi- line compression transfer function including one or more compression thresholds that are derived from a fractional measure of a number of frames of the audio track at one or more predetermined levels.

However Kitamura teaches that compressing using the actual compression scheme comprises compressing with a multi-line compression transfer function (reads on, multichannel) derived from the statistical frequency data, the multi- line compression transfer function including one or more compression thresholds that are derived from a fractional measure (by compared) of a number of frames(reads on, number of samples)

of the audio track at one or more predetermined levels (see figs. 1-4 and col.5 line 41-col. 6 line 67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to combine the teaching of Kitamura into Saunders to provide a preferred equalization requirement and automatically adjust equalization of the audio and to provide a proper multichannel audio output for the sound system.

Regarding Claim 19, Saunders discloses obtaining a mean loudness deviation value from the loudness values for the plurality of audio frames; determining a test mean loudness deviation value from the test statistical frequency data; and comparing the mean loudness deviation value and the test mean loudness deviation value with a desired mean loudness deviation value when deriving the actual compression scheme (Figs. 1,2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34).

Regarding Claim 20, Saunders discloses a method of processing an audio track comprising: obtaining statistical frequency data for the audio track (Figs. 1, 5, and 13-14; column 17, line 16 to column 18, line 34); applying a compression scheme to the statistical frequency data to obtain an estimate of statistical frequency data that would result from applying the compression scheme directly to the audio track (Figs. 1, 2A-2B, 3, 5, and 13-14; column 8, line 7 to column 9, line 51; column 17, line 16 to column 18, line 34); determining an estimated overall compressed loudness value from the estimate of statistical frequency data (Figs. 1, 2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34); compressing the audio track using the compression scheme to obtain a compressed audio track (Figs. 1, 2A-2B, 3, 5, and 13-14; column 17, line 16 to

column 18, line 34); and applying a gain to the compressed audio track based on a comparison between the estimated overall compressed loudness value and a desired loudness value (Figs. 1, 2A-2B, 3, 5, and 13-14; column 17, line 16 to column 18, line 34); wherein compressing using the compression scheme comprises compressing with a multi- line compression transfer function (reads on PCPV/PCA and/or SCRA) derived from the statistical frequency data, the compression transfer function including one or more compression thresholds (reads on, 8:1) that are derived from a fractional measure of the audio track at one or more predetermined levels (see Figs. 1, 5, and 13-14; and see column 13, line 8 to column 14, line 10; column 17, line 16 to column 18, line 34; column 23, lines 48-67); but Saunders does not explicitly teach the compression transfer function including one or more compression thresholds that are derived from a fractional measure of a number of frames of the audio track at one or more predetermined levels.

However Kitamura teaches that compressing using the compression scheme comprises compressing with a multi- line compression transfer function(reads on, multichannel) derived from the statistical frequency data, the compression transfer function including one or more compression thresholds that are derived from a fractional measure(by compared) of a number of frames(reads on, number of samples) of the audio track at one or more predetermined levels (see figs. 1-4 and col.5 line 41- col. 6 line 67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to combine the teaching of Kitamura into Saunders to provide

a preferred equalization requirement and automatically adjust equalization of the audio and to provide a proper multichannel audio output for the sound system.

Regarding Claim 21, Saunders discloses the overall compressed loudness value is obtained by: obtaining a plurality of individual loudness values from the estimate of statistical frequency data; applying a weighting factor to each of the individual loudness values to obtain weighted loudness values; and aggregating the weighted loudness values to obtained the overall compressed loudness value for the audio track (Figs. 1, 2A-2B, 3, 5, and 13-14; column 8, line 7 to column 9, line 51; column 17, line 16 to column 18, line 34; column 23, lines 48-67; column 22, lines 53-60; column 26, line 40 to column 4).

Regarding Claim 17, Saunders discloses a metadata which is the audio control information, but does not expressly disclose the weighted loudness values of the plurality of audio frames are aggregated using a histogram (column 7, line 42 to column 8, line 60). However, the examiner takes Official Notices that it well known in the art to provide metadata comprises histogram associated with the audio control information in order to further enhance the playback features available. Therefore it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Saunders to have the metadata of Saunders, comprise histogram associated with the audio control information in order to further enhance the playback features available.

4. Claim 4-13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders (US PAT. 6,351,733) as modified by Kitamura (US PAT. 6,704,421) as applied to claims 1-4 above, and further in view of Nakano (US PAT. 5,404,315).

Regarding 4, Saunders as modified by Kitamura does not explicitly teach the step of deriving the time varying gain comprises:

deriving, from histogram data of levels encountered in the audio track, an original dynamic spread value representing a spread of the levels encountered in the audio track;

performing a comparison between the original dynamic spread value and a desired dynamic spread value; and deriving a parameter for the transfer function from the comparison.

However, Nakano teaches that the step of deriving the time varying gain comprises:

deriving, from histogram data of levels encountered in the audio track, an original dynamic spread value representing a spread of the levels encountered in the audio track (see col. 7 lines 42-51);

performing a comparison between the original dynamic spread value and a desired dynamic spread value (see col. 10 line 41-59); and deriving a parameter (reads on coefficient) for the transfer function (equation 1) from the comparison (see figs 3-4 and 6 and see col. 10 lines 41-59).

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to combine the teaching of Nakano into Saunders and Kitamura to provide better gain control for enhancing the audio output to the user.

Regarding Claim 5, Nakano discloses deriving parameters comprises: determining a slope of a segment of a compressor transfer function (column 6, line 5 to column 7, line 37); and determining a threshold between two segments (a, b, c) of the compressor transfer function (formula 1 and see figs 3-4, 6 and col. 6 line 5-col. 7 line 37 and discussion above claim 4). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 6, Nakano discloses determining the slope comprises: applying a test compression scheme to the histogram data to obtain test histogram data, the test compression scheme including a test slope (Figs. 1, 7-10; column 7, lines 42-51); determining a test dynamic spread value from the test histogram data (column 7, line 42 to column 8, line 60); and deriving the slope based on a comparison of the original dynamic spread value, the desired dynamic spread value and the test dynamic spread value (column 6, lines 5-27). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 7, Nakano discloses the slope for the compressor transfer function is determined using interpolation (column 6, line 5-27). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 8, Nakano discloses the slope for the compressor transfer function is determined using iteration (column 7, line 42 to column 8, line 60). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 9, Nakano discloses the original dynamic spread value is derived from a mean absolute deviation from a mean loudness value for the audio track (column

4, line 57 to column 5, line 3). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 10, Nakano discloses the original dynamic spread value is derived from a mean absolute deviation from a median loudness value for the audio track (column 4, line 57 to column 5, line 3). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 11, Nakano discloses the parameters include a level of a threshold separating two segments of a compressor transfer function (Figs. 1, 7-10; column 10, lines 41-59). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 12, Nakano discloses specifying a fraction representing a proportion of the audio track to which compression will be applied (column 6, lines 5-27); deriving from the histogram data a loudness value corresponding to a point above or below which the fraction of the histogram data is located; and using the loudness value as a threshold separating two segments of a compressor transfer function (Figs. 1, 7-10; column 7, line 42 to column 8, line 60; column 10, lines 41-59). Note the discussion of claim 4 for a motivation to combine.

Regarding Claim 13, Nakano discloses deriving a test overall loudness value from the test histogram data; deriving a fixed post-gain value from the test overall loudness value and from a desired loudness value; and applying the time varying gain and the fixed post-gain value to the audio track (Figs. 1, 7-10; column 6, lines 5-27; column 7, line 42 to column 8, line 60). Note the discussion of claim 4 for a motivation to combine.

Response to Arguments

5. Applicant's arguments with respect to claims 1-21 have been considered but are moot in view of the new ground(s) of rejection.

***Conclusion***

6. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

8. Any response to this action should be mailed to:

Mail Stop ____(explanation, e.g., Amendment or After-final, etc.)

Commissioner for Patents
P.O. Box 1450
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Facsimile responses should be faxed to:
**(571) 273-8300**

Hand-delivered responses should be brought to:
Customer Service Window
Randolph Building
401 Dulany Street
Alexandria, VA 22314

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Lao, Lun-See whose telephone number is (571) 272-7501 The examiner can normally be reached on Monday-Friday from 8:00 to 5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Vivian Chin, can be reached on (571) 272-7848.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Technology Center 2600 whose telephone number is (571) 272-2600.

Lao, Lun-See
/Lun-See Lao/
Examiner, Art Unit 2615
Patent Examiner
US Patent and Trademark Office
Knox
571-272-7501

Date 10-16-2008

/Vivian Chin/

Supervisory Patent Examiner, Art Unit 2614